Issues on Dummy-Head HRTFs measurements

Toledo, Daniela; Møller, Henrik

Published in:
Audio Engineering Society Convention Papers

Publication date:
2009

Audio Engineering Society
Convention Paper 7727
Presented at the 126th Convention
2009 May 7–10 Munich, Germany

Issues on Dummy-Head HRTFs measurements

Daniela Toledo[1] and Henrik Møller[1]

[1] *Section of Acoustics, Aalborg University, 9220, Aalborg-Ø, Denmark*

Correspondence should be addressed to Daniela Toledo (dt@es.aau.dk)

ABSTRACT

The dimensions of a person are small compared to the wavelength at low frequencies. Therefore, at these frequencies head-related transfer functions (HRTFs) should decrease asymptotically until they reach 0 dB -i.e. unity gain- at DC. This is not the case in measured HRTFs: the limitations of the equipment used result in a wrong -and random- value at DC and the effect can be seen well within the audio frequencies. We have measured HRTFs on a commercially available dummy-head Neumann KU 100 and analyzed issues associated to calibration, DC correction and low frequency response. Informal listening tests suggest that the ripples seen in HRTFs with a wrong DC value affect the sound quality in binaural synthesis.

1. INTRODUCTION

Head-related transfer functions (HRTFs) contain the transformations that affect the sound in its path to the eardrum of a listener. As a contribution to a current round robin of HRTFs measurement systems [1], we have measured HRTFs of the dummy-head Neumann KU 100 for 85 directions in an anechoic chamber. These HRTFs measurements are used in the present investigation as a case study to approach different issues such as calibration, DC correction and low frequency response.

Firstly, as part of this introduction some basic concepts such as HRTFs, free field transfer function and diffuse field equalization are reviewed. The theoretical framework for HRTFs measurements is also introduced. In the second section, we present the generalities of the HRTFs measurement procedure that was followed in this investigation. The issues that are further analyzed are mentioned in their context. In following sections, the issues of calibration, DC correction and low frequency response are presented. These issues are not independent from each other but will be treated separately for simplicity. The Discussion follows where the relationship among the issues is covered. Finally, some concluding remarks are made.

1.1. Background

The Neumann KU 100 is a dummy-head with built-in diffuse field equalization (see the documentation of the dummy-head in [2]). Therefore, some basic concepts are reviewed in the following -e.g. HRTF, free field transfer function, diffuse field equalization, etc. Even though these definitions are clearly stated in the literature -for example [3][4][5]- it seems that they are not always followed.

If the human anthropometry is considered as a linear time-invariant system, the transformations that it imposes over an impinging sound can be expressed as a transfer function. By definition, two terms are then necessary to obtain the transfer function: the output and the input to the system. In the broad sense, HRTFs are defined as the complex pressure division of the sound incoming at the ears of a subject ($P2$ or output of the system) to the sound at the position of the center of the head when the subject is absent ($P1$ or input to the system). This can be expressed as:

$$HRTF = \frac{P2}{P1} \tag{1}$$

As one $P2$ measurement exists for each ear, HRTFs are defined in pairs which are angle dependent. It is normal practice to sate to which coordinate system the HRTFs are referred to. Angles are usually given in (azimuth ϕ, elevation θ), which is the nomenclature also used in this work.

The microphone position for the measurement of $P2$ can vary: it can be at the entrance of the blocked or open ear canal, at the eardrum or at some known position in the ear canal. A review on the choice of measurement point is given in [6].

The definition of HRTF as in Eq.1 has received other names in the literature: free field transfer function [3], external ear transfer function [7], transfer function from free sound field to ear canal entrance -or to the eardrum- [7],directional transfer function [8], among others.

Other transfer functions were defined by Blauert in [3]: interaural transfer function and monaural transfer function. The former relates the sound pressures measured at both ears of the subject, where the reference sound pressure is that at the ear facing the sound source. Monaural transfer functions relate the sound pressure at the ears of a subject to sound pressure measured at the same position but with the sound source located at a reference position -as a rule, it corresponds to the position to the front with coordinates $(0°, 0°)$. Monaural transfer functions can also be referenced to diffuse field [4]. In that case, the reference is the average of the transfer functions from all directions.

If Eq.1 is considered from a practical point of view, $P2$ and $P1$ are ideal transfer functions that have to be obtained from real measurements. These are M_{P2} and M_{P1} measurements respectively, which also contain the transfer function of the measurement setup. In the case presented here, these include the transfer functions of a computer-based mls system [9], an RME ADI-8 DS AD/DA converter, a Pioneer A-616 power amplifier, 3 inch loudspeakers VIFA M10MD-39, the transfer function of the microphones used for M_{P2} and M_{P1} measurements, and finally the transfer functions of the Brüel & Kjær 2607 measuring amplifiers.

$$HRTF = \frac{M_{P2}}{M_{P1}} \tag{2}$$

If the same setup is used for both M_{P2} and M_{P1} measurements, the transfer functions mentioned above are canceled out in Equation 2. That is the case in the presented investigation, except for the transfer function of the microphones. Following a requirement of the round robin for which these measurements were done, the internal microphones of the dummy-head Neumann KU 100 were used for M_{P2} measurement. The pressure field microphone Brüel & Kjær 4136 was chosen for the reference measurement M_{P1}. Therefore, HRTFs are obtained by compensating for the transfer characteristics of the microphones used:

$$HRTF = \frac{M_{P2} \cdot H_{mic.P1}}{M_{P1} \cdot H_{mic.P2}} \Rightarrow HRTF = \frac{H_{P2}}{H_{P1}} \tag{3}$$

This is the same as Eq.1:

$$HRTF = \frac{P2}{P1} = \frac{H_{P2}}{H_{P1}} \tag{4}$$

2. HRTFS MEASUREMENT PROCEDURE

The details of the conducted HRTFs measurements are given in this section. The issues that will be developed in further sections are also pointed.

2.1. Measurement Setup

The measurements were made in an anechoic chamber. The dummy-head stood in the center of an arc and 15 loudspeakers were placed along it with 22.5° of separation. The distance from the loudspeaker to the point in the center of the head was 1.5 m. Sound sources were 3 inch VIFA M10MD-39 loudspeakers mounted in hard plastic balls.

The head was rotated in 30° steps by means of a turntable Brüel & Kjær type 3921. The rotation was done with respect to the head's stand, which was not coincident with the vertical axis crossing the center of the head -angular and distance errors were introduced by this procedure, being the maximum angular error equal to -1° and the maximum distance error equal to ± 5 cm. A total of 85 HRTFs were measured.

A two-channel computer-based MLS system [9] was used for transfer function measurements. The computer was equipped with a digital sound card RME HDSP 9632 and generated digital signals that were fed to an RME ADI-8 DS AD/DA converter. Analog signals were then fed to a power amplifier Pioneer A-616 calibrated to provide 0 dB gain. The output of the power amplifier was sent to a switch box, controlled through the parallel port of the PC, which diverted the signal to the desired sound source. The balanced 5-pin XLR output of the dummy-head was used to provide external polarization -a phantom power supply Neumann BS 48-i2 was used- and to obtain the output signals. Internal microphones were calibrated for their sensitivity at 1 kHz (see Section 3). The balanced outputs were converted into unbalanced and delivered to two measuring amplifiers Brüel & Kjær 2607. As these measuring amplifiers inverted the phase of the signals, their transfer function was deconvolved from the measurements of $P2$ to obtain the correct phase response (see 2.2). In the case of $P1$ measurements, only the gain factor of the measuring amplifier was accounted for. The output from the measuring amplifiers fed the signals to the RME ADI-8 DS AD/DA converter. Digital signals went back to the PC for the transfer function computation. The results were impulse responses of 2048 samples length, at a sampling frequency of 48kHz.

2.2. Frequency response of the setup

The assumption of a flat frequency response and deviations from nominal gains were verified with measurements. The Pioneer A-616 power amplifier presented negligible deviations from the nominal 0 dB gain. The measuring amplifiers showed deviations from the nominal gain settings of the order of 0.2 dB and they were compensated for all measurements in the post-processing stage. In the case of $P2$ measurements, the whole transfer function of the measuring amplifiers was deconvolved. This procedure corrected a phase inversion introduced by the measuring amplifiers. In the case of $P1$ measurements, the phase needed not to be corrected: both the microphone Brüel & Kjær 4136 and the measuring amplifier produce a phase inversion, compensating each other.

Calibration of the internal Neumann KU 100 microphones was done with a sound level calibrator Brüel & Kjær 4230. It was not a straightforward procedure, as it is explained below in Section 3. Since measurements of $P2$ were done over two different days, two calibration values were obtained for each internal microphone.

The reference microphone Brüel & Kjær 4136 ($\frac{1}{4}$ inch microphone) has a flat frequency response in the range 20Hz-40kHz, and was also calibrated for its sensitivity at 1kHz.

2.3. Post-processing

Original measurements had 2048 data points. A rectangular window was applied to the raw $P1$ and $P2$ measurements and data points from 55 to 310 were used. This window ensured that all the impulse responses had died out, but could not exclude the first reflections from the setup. The files were further processed to account for the gain and phase of the measuring amplifiers and the sensitivity of the microphones. The frequency response of the measuring amplifier was deconvolved from $P2$ measurements as mentioned in Section 2.2 -this was done by a division in the frequency domain.

The low frequencies of $P1$ and $P2$ measurements presented different transfer characteristics, and it was hypothesized that there was a gain at low frequency in the Neumann KU 100 internal microphones. In order to confirm this, the low frequency investigation reported in Section 5 was conducted. As a result of that investigation, $P2$ measurements were filtered with inverse filters that equalized the low frequency response of the internal microphones.

2.4. Computation of HRTFs

Once $P2$ measurements were filtered, the free field HRTFs were computed as a complex pressure division (division in the frequency domain) according to Equation 4. The results were low pass filtered. HRIRs were computed from the inverse Fourier Transform. HRIRs were circularly shifted 60 samples to ensure casualty. All HRIRs were shifted the same amount of samples in order to keep the interaural time difference information. Finally, HRIRs were DC corrected in time domain to provide a meaningful value at 0 Hz and minimize the effects of truncation, as explained in Section 4.

3. CALIBRATION

In the context of microphones, the amount of electrical output for a certain amount of sound pressure presented to a microphone is expressed as sensitivity. The units commonly used are V/Pa (volts output per Pascal of pressure applied) or dB re. 1V/Pa (decibels relative to 1 volt per Pascal). Microphone sensitivities are determined through calibration. Calibration can be performed either in the field or in a laboratory, and there are several methods that can be used -comparison method, substitution method, calibration by the use of a pistonphone or sound level calibrator, among others. The interested reader is referred to [10] for a review of different methods.

The requirement of calibration is widely accepted since it ensures that measurements are correctly done and the equipment involved is accurate. It also accounts for environmental variabilities, enabling measurements to be compared. This is a critical issue in the context of HRTFs, which result from the ratio of two measurements. As explained in 1.1, if the microphones used for $P1$ and $P2$ measurements are not the same, their sensitivities and frequency response will be different. It can be the case that, even if the same microphone is used, the sensitivity changes due to environmental conditions. Therefore, proper calibration has to be conducted in order to cancel the $H_{mic.P2}$ and $H_{mic.P1}$ terms as in Eq.3, unless they are equal.

Even though the reviewed concepts are well established and are considered as normal procedure, a proper calibration is not always straightforward. This was the case when attempting to calibrate the internal microphones of the Neumann KU 100 dummy-head. In the following, HRTFs obtained with different calibration values are compared.

3.1. Neumann KU 100 internal microphones

The internal microphones of the dummy-head under study consisted of two pressure transducers with nominal sensitivity at 1kHz of 20mV/Pa ± 1 dB. The microphone capsules were hosted in ear adapters that contained ear channels. The ear adapters were attached to a cylindrical enclosure that included built-in filters. Since the microphone capsules could not be detached from the ear adapters nor the cylindrical enclosure, only calibration with a sound level calibrator was possible. The manufacturer of the dummy-head recommends the Neumann PA100 adapter for calibration, which is an accessory part to the dummy-head. That adapter fits into a $\frac{1}{2}$ inch Brüel & Kjær adapter, which in turn fits into a 1 inch Brüel & Kjær calibrator. However, the Neumann PA100 was not available when the measurements were done. An alternative calibration procedure was followed.

3.2. Calibration Procedure

The calibration procedure started on the day of the measurements, when the sensitivity of the microphones was approximated by combining a $\frac{1}{2}$ inch Brüel & Kjær adapter with a $\frac{1}{4}$ inch Brüel & Kjær adapter. The insertion of the Neumann KU 100 internal microphones in the $\frac{1}{4}$ inch adapter was sealed to avoid leakages -since they were slightly smaller than $\frac{1}{4}$ inch.

The Neumann PA100 adapter was received at a later day and used to verify the calibration done the day of the measurements. The difference between the two calibration procedures -i.e. ($\frac{1}{2}$ inch Brüel & Kjær adapter + $\frac{1}{4}$ inch Brüel & Kjær adapter) vs. ($\frac{1}{2}$ inch Brüel & Kjær adapter + Neumann PA100 adapter) was computed. In average, the difference between the two calibration procedures amounted to 2 dB. This difference includes an insertion loss of 1.7dB reported by the manufacturer of the Neumann PA100.

3.3. Results

Figure 1 shows the measured HRTFs for direction $(0°, 0°)$, where $P1$ and $P2$ have not been post-

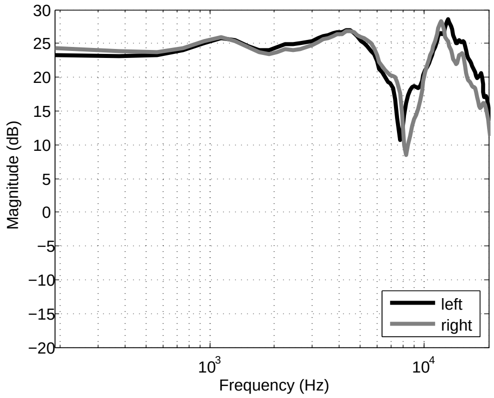

Fig. 1: Measured HRTFs for direction (0°, 0°), without microphone calibration.

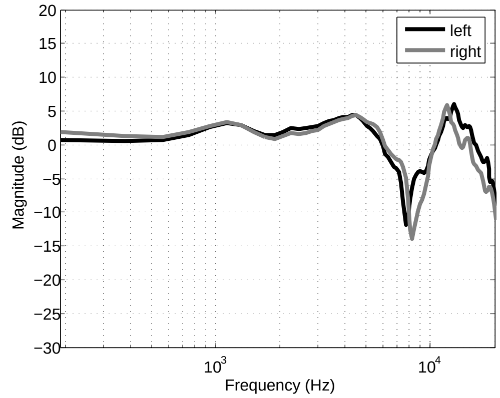

Fig. 2: Same HRTFs as in Fig.1 but with microphone calibration. Microphones used for $P2$ measurements were calibrated by combining a $\frac{1}{2}$ inch with a $\frac{1}{4}$ inch Brüel & Kjær adapters.

processed with their corresponding calibration values. It can be seen that the whole frequency response is shifted upwards, and they do not decrease asymptotically until reaching 0 dB at DC. Figure 2 shows the processed HRTFs where both $P1$ and $P2$ have been calibrated with the values obtained by combining a $\frac{1}{2}$ inch Brüel & Kjær adapter with a $\frac{1}{4}$ inch Brüel & Kjær adapter. It can be seen that the response is still shifted upwards, meaning that there is still an added gain factor that should be accounted for.

Figure 3 shows the same HRTFs as in Figure 2, but the calibration of $P2$ measurements have been corrected by 2 dB according to the findings mentioned before (difference between a $\frac{1}{4}$ inch Brüel & Kjær adapter and the Neumann PA100 adapter). It can be seen that the low frequency response is closer to the expectation but there are still some differences between right and left side. Furthermore, there are deviations from the frequency response reported by the manufacturers for the same direction [2].

The calibration procedure described is not free from errors: they could arise as a result of the uncertainty in the calibration method (0.07 to 0.3dB according to [10]) and also from the calculation of the difference between the two calibration procedures -with and without the Neumann PA100 adapter. However, it was hypothesized that the low frequency behavior seen in Figure 3 was not due to calibration errors but due to the characteristics of the Neumann KU 100 internal microphones. An investigation was conducted and is reported in Section 5.

4. DC CORRECTION

At low frequencies, the dimensions of a person become much smaller than a wavelength. Hence, ideal HRTFs are expected to decrease asymptotically un-

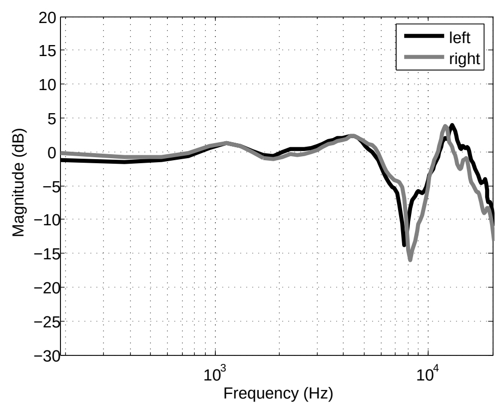

Fig. 3: Same HRTFs as in Fig.2 but the calibration of the microphones used for $P2$ measurements were corrected by 2 dB -which was the average difference between using a $\frac{1}{4}$ inch Brüel & Kjær adapter and the Neumann PA100 adapter.

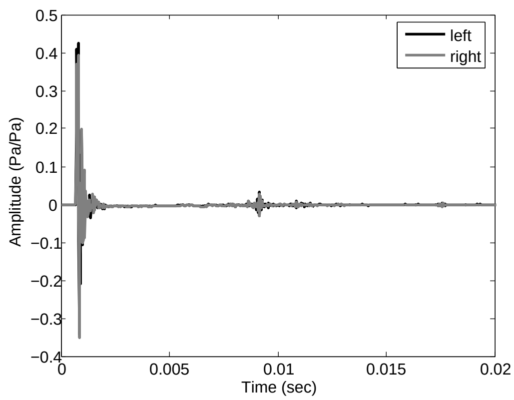

Fig. 4: Impulse responses of the FIR filters of HRTFs corresponding to direction $(0^\circ, 0^\circ)$. Filters were constructed from 1024 samples of the HRIRs measurements -hence including reflections from the setup. Impulse responses were computed with a length of 4096 samples at a sampling frequency of 48kHz.

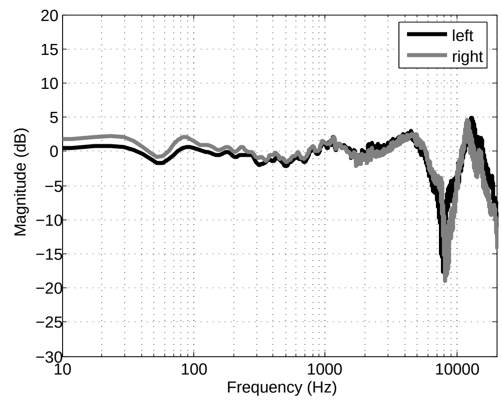

Fig. 5: Frequency responses of the FIR filters shown in Fig.4.

til they reach 0 dB or unity gain at DC. This is not the case in measured HRTFs mainly due to two factors: limitations of the measurement setup and restrictions on the length of the HRTFs filters.

4.1. Limitations of the measurement setup

Sound is not reproduced nor measured at DC and this holds for both $P1$ and $P2$ measurements. If values are obtained at DC in these measurements, they obey the offset voltage properties of the acquisition equipment used. This issue has already been pointed in [5][11]. Moreover, the DC value in HRTFs results from the ratio of two measurements with non-zero DC value. The ratio, therefore, results in a meaningless -wrong and more or less random- value at DC.

4.2. Length of the HRTFs filters

The length of measured HRIRs is often decided so as to avoid possible reflections from the setup. These short HRIRs measurements are often implemented as FIR filters which seem to be perceptually valid: in previous experiments at our laboratory [12] it has been shown that HRIRs as FIR filters of 72 taps of length (sampled at 48 kHz) were long enough to convey all the needed cues to sound localization. Nevertheless, these short filters impose a poor frequency resolution and noticeable consequences are seen in the low frequency range -which is then represented by too few taps. For example, Figure 4 shows the impulse responses of two FIR filters of 1024 taps, corresponding to direction $(0^\circ, 0^\circ)$. Some reflections can be clearly seen around 0.01 seconds, which affect the whole frequency responses as seen in Figure 5. In turn, Figure 6 shows the obtained impulse responses if the FIR filters are constructed with only 256 samples of the HRIR measurements. The corresponding frequency responses of the filters are shown in Figure 7. Even though the responses are smoother than in Figure 5 due to the lack of reflections, the low frequencies are farther from 0 dB than in Figure 5. Moreover, some ripples can be seen in the low frequency range -around and above 200Hz.

The plotted responses in Figures 4 to 7 are 4096 samples long but the FIR filters from which they were obtained are much shorter. Even though the FIR filters are determined for a few limited frequencies, they are still filtering those frequencies in-between and unfortunately ripples appear. Informal listening tests suggest that the ripples seen in the low frequency range affect the sound quality in binaural synthesis, as already reported in [5].

4.3. Results

In Figure 8, the DC value of the filters was corrected to equal unity gain. This was done in the time domain, by ensuring that the sum of all taps equals 1. This procedure accounts for the two aforementioned

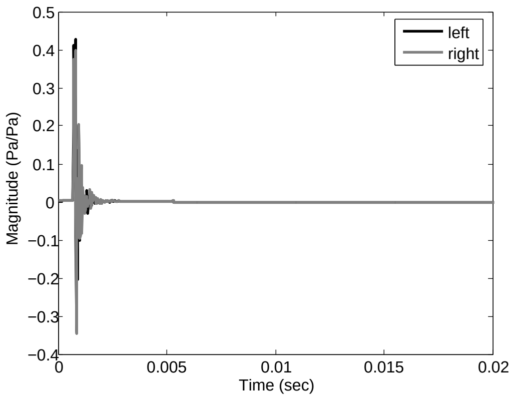

Fig. 6: Same as in Fig.4, but FIR filters were constructed from 256 samples of the HRIRs measurements -therefore avoiding some reflections from the setup.

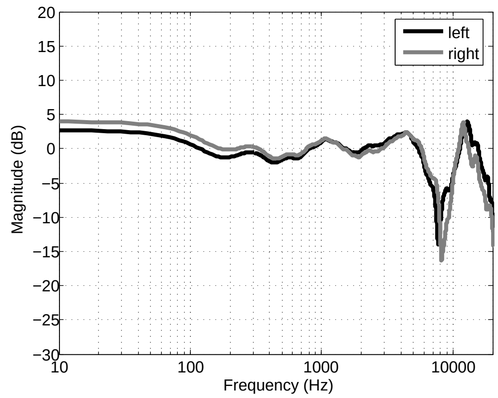

Fig. 7: Frequency responses of the FIR filters shown in Fig.6.

problems: limitations of the setup and low frequency ripples due to the length of the filters.

Regarding the limitations of the setup, correcting DC ensures that HRTFs asymptotically reaches 0 dB at DC -which is a theoretically valid procedure and gives a meaningful value at 0Hz.

Regarding the low frequency ripples, it can be seen in Figure 8 that they are minimized by controlling DC. Figure 9 shows a zoom in the low frequency range were the ripple control can be seen more clearly. Informal listening tests also showed an improvement in the perceived quality of the signals synthesized with such corrected HRTFs. It has to be pointed, however, that the quality problem was only seen when large differences were present between the DC values at both ears. If the interaural difference at DC is small, and both values are around 0 dB, the quality does not seem to be affected.

5. LOW FREQUENCY COMPENSATION

In our experience, the combination of a proper calibration and DC control ensures a well-behaved HRTF in the low frequency range. That means that HRTFs decrease asymptotically until reaching 0 dB at DC, as we have stated throughout this work. Moreover, in the median plane both left and right signals are expected to be equal at low frequencies -disturbances due to positioning errors or asymmetries are possible but at higher frequencies. The Neumann KU 100 dummy-head, however, presented differences in the low frequency range -see Figure 8 and the signal differences in the range up to 500Hz- possibly due to the diffuse field equalization filters. As the impossibility of proceeding with a calibration by the method of substitution or comparison -see Section 3- made the frequency response of the

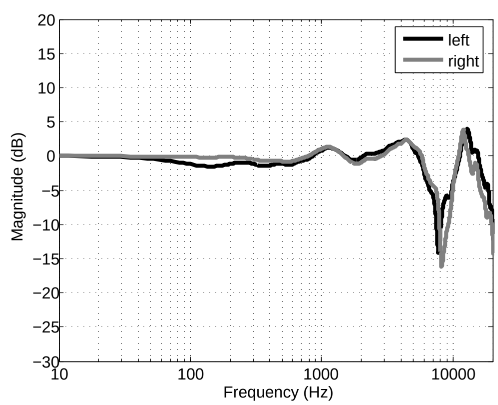

Fig. 8: Same as in Fig.7, but the DC value of the filters was corrected to equal unity gain. This was done in the time domain, by ensuring that the sum of all taps equals 1.

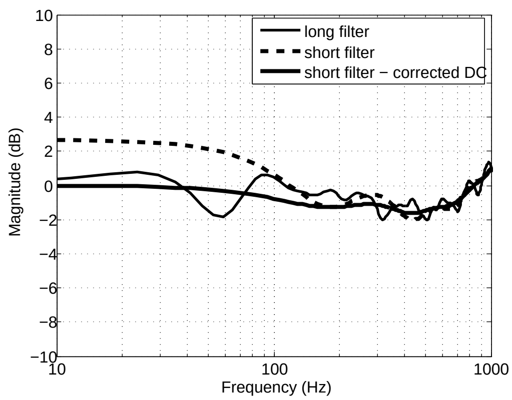

Fig. 9: Comparison of the low frequency response of the FIR filters with and without the DC value corrected, corresponding to the left side HRTF for direction $(0°, 0°)$.

microphones unknown, an alternative procedure was followed to investigate the response of the Neumann KU 100 dummy-head at low frequencies.

5.1. Measurement of the low frequency characteristics of the Neumann KU 100 internal microphones

The low frequency characteristics of the Neumann KU 100 internal microphones were investigated by attaching two reference microphones Brüel & Kjær 4193 modified with UC 0211 capsules. These microphones had a flat frequency response from 0.07Hz to 20kHz and were calibrated to their sensitivity at 1kHz. The head with the attached microphones was put inside a sealed loudspeaker cabinet. A loudspeaker SEAS 33 F-WKA and a 40cm x 40cm x 40cm cabinet were used. The sound pressure inside a sealed cabinet is proportional to the displacement of the cone at very low frequencies -i.e. until the resonance of the system- and high sound pressure levels are reproduced. Outside the cabinet, the pressure is proportional to volume acceleration at those very low frequencies -it increases 12dB per octave and very low sound pressure levels are reproduced. Therefore, all the measurements described in this section were done with the dummy-head inside the sealed cabinet.

The frequency characteristics of the Neumann KU 100 internal microphones were determined from 2Hz to 20Hz with a 1Hz resolution and from 20Hz to 315Hz at the center frequencies of standard 1/3 octave bands [13]. A sine wave generator Brüel & Kjær 1027 was used to generate signals at each frequency of interest. For each of these sine waves, the voltage registered by the measuring amplifiers at the output of the microphones was recorded. This was done for both reference microphones and Neumann KU 100 internal microphones consecutively, without any change in the setup. These measurements were repeated in different days.

Since the reference microphones presented a flat frequency response down to 0.07Hz, the difference in dB between the reference and internal microphones equals the frequency characteristics of the Neumann KU 100 internal microphones.

5.1.1. Results

The frequency characteristics of the Neumann KU 100 internal microphones are shown in Figure 10, normalized to their sensitivity at 1kHz. It can be seen that the frequency characteristics of both left and right microphones are very different at very low frequencies. The left side frequency response complies with the specifications provided by the manufacturer (high pass filter with cut-off frequency at 8Hz), even though there is a small gain above 10Hz. The right side frequency response, however, is far from the specifications.

The Neumann KU 100 dummy-head has options for high pass filtering with a cut-off frequency of 40Hz and 140Hz. Measurements were also conducted with these settings and the results are shown in Figure 11. By inspection of Figures 10 and 11, a gain in the right internal microphone can be seen with respect to the left internal microphone. This explains the differences seen between left and right signals in previous figures -for example, see Fig.7.

From these figures, it can be concluded that the Neumann KU 100 has a low frequency response that, if excited, will produce a long impulse response. However, these frequencies are only excited if a loudspeaker that reproduce sound at those frequencies is used for the measurements. This is not the case for the VIFA loudspeakers used for $P1$ and $P2$ measurements presented in previous sections. If a loudspeaker with a better low frequency response had

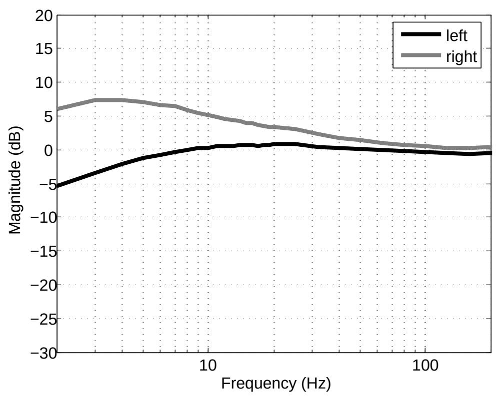

Fig. 10: Low frequency response of the Neumann KU 100 internal microphones normalized to their 1kHz sensitivity.

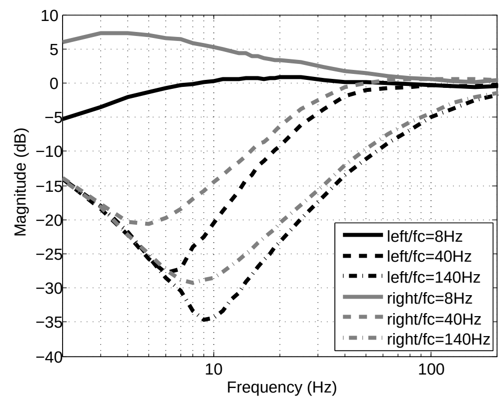

Fig. 11: Low frequency response of the Neumann KU 100 internal microphones normalized to their 1kHz sensitivity, with built-in high-pass filters applied. (Note change of scale with respect to Fig.10)

been used, the measurements would have gone much further in time than the reflections of the setup. Moreover, the results would have been meaningless, since at low frequencies the HRTFs should present asymptotic behavior towards DC. We hypothesize that the low frequency characteristics seen are a result of the diffuse field built-in circuits of the Neumann KU 100 dummy-head.

5.2. Low frequency range control

After investigating the low frequency characteristics of the Neumann KU 100 internal microphones, it was decided to control the low frequency range of all $P2$ measurements by filtering. From the frequency responses shown in Fig.10, inverse filters were constructed. The responses in frequency were completed in 1Hz steps. In the range from 20Hz to 1kHz, the values were linearly interpolated between actual measured ones. Above 1kHz, the responses were set to unity gain. From the obtained frequency response, linear phase FIR filters were computed by the windowing method. The minimum-phase representation of these filters was obtained by the Hilbert Transform [14] [15]. The minimum-phase filters were truncated to 256 taps and the inverse was computed. The results were implemented as FIR filters and applied to all $P2$ measurements.

5.2.1. Results

After filtering all $P2$ measurements with the aforementioned FIR inverse filters, the HRTFs were computed again. The results for the direction $(0°, 0°)$ are shown in the Figure 12. If compared with Figure 3 of Figure 8, it can be concluded that the low frequencies of the HRTFs are as expected for a direction in the median plane.

6. DISCUSSION

The three issues examined in previous sections have

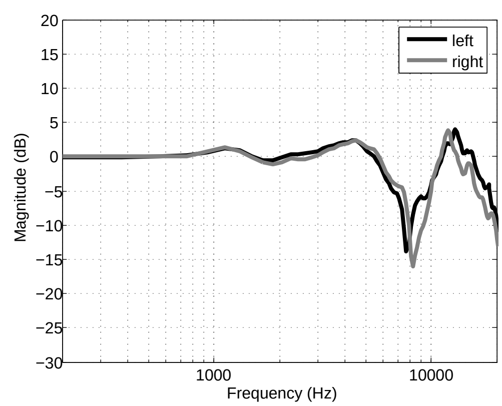

Fig. 12: Same as Fig.8, but low frequencies have been equalized to account for the characteristics shown in Fig.10

been presented alone even though they are related among each other.

Calibration of the microphones are a requirement of any acoustical measurement. In the context of HRTFs, calibration ensures a correct gain. It is not straightforward to assess calibration errors at high frequencies in HRTFs: in this range, they present much inter-subject variability that is furthermore direction dependent. However, errors become self evident by inspection of the low frequency range: HRTFs are expected to reach 0 dB at 0 Hz, asymptotically. Deviations from this behavior -for example, in Figures 1 and 2- can be hypothesized as closely related to poor or inexistent calibration. In the ideal case, a proper calibration accounts for the whole frequency response of the microphones if they are non-flat. This is possible if microphones are calibrated by methods such as comparison or substitution. An alternative is to investigate particular frequency ranges that deviate from a flat response, as in the reported low frequency investigation. The case of the Neumann KU 100 is unlike others, however, as measurement microphones usually present a flat frequency response in the whole range of audio frequencies.

Once HRTFs present the expected gain and an asymptotic decrease toward DC, it is only the value that DC takes which is meaningless. Therefore, proper calibration is also required for a valid DC correction. For example, controlling DC for the responses in Figure 3 gives correct results (after controlling the low frequency range, it gives Fig.12), but controlling DC for the responses in Figure 1 would create a wrong jump of 25 dB between DC and the next frequency component.

As mentioned earlier in this work, HRTFs are often implemented as short FIR filters which convey all the necessary localization cues. However, short FIR filters define low frequencies with too few frequency components. The frequencies in-between those that are defined, are not controlled. Ripples appear in those frequencies in-between, as shown in Figure 9. One possibility of controlling the ripples would be to make longer measurements - then, more frequency points would be controlled. However, longer impulse responses require more demanding reflection-free setups and loudspeakers that can reproduce sound at those low frequencies. The procedure becomes troublesome, particularly when considering that the ripples are meaningless since HRTFs should decrease asymptotically. DC correction is a much more convenient way of minimizing those ripples which allows using loudspeakers with shorter impulse responses.

7. CONCLUSION

A case study of dummy-head HRTFs measurements was presented to discuss the requirements of a proper calibration, DC correction and low frequency control. These are necessary conditions to ensure correct HRTFs: they should decrease asymptotically until they reach 0 dB at DC, preserving the audio quality. In our investigation, it was seen that the three issues presented are connected: a proper calibration ensures a correct measurement at low frequencies and makes DC control a valid procedure, and DC control ensures a meaningful value at 0 Hz apart from minimizing low frequency ripples.

8. ACKNOWLEDGMENTS

The authors would like to thank Brian F.G. Katz and Durand R. Begault for inviting our laboratory to contribute to the round robin of HRTF measurement systems, and providing the Neumann KU 100 dummy-head.

9. REFERENCES

[1] B. F. G. Katz and D. R. Begault, "Round robin comparison of hrtf measurement systems: preliminary results," in *Proceedings of the 19th International Congress on Acoustics*, 2007.

[2] http://www.neumann.com/?lang=en&id=current_microphones&cid=ku100_publications.

[3] J. Blauert, *Spatial Hearing. Revised Edition.* MIT, 1997.

[4] H. Møller, "Fundamentals of binaural technology," *Applied Acoustics*, vol. 36, no. 3/4, pp. 171–218, 1992.

[5] D. Hammershøi and H. Møller, *Binaural technique - Basic Methods for Recording, Synthesis and Reproduction. Chapter in book Communication Acoustics.* Springer-Verlag, ISBN 3-540-22162-x, 2005.

[6] D. Hammershøi and H. Møller, "Sound transmission to and within the human ear canal," *The Journal of the Acoustical Society of America*, vol. 100, no. 1, pp. 408–427, 1996.

[7] S. Mehrgardt and V. Mellert, "Transformation characteristics of the external human ear," *Journal of the Acoustical Society of America*, vol. 61, no. 6, pp. 1567–1576, 1977.

[8] J. C. Middlebrooks and D. M. Green, "Sound localization by human listeners," *Annual Review of Psychology*, vol. 42, pp. 135–159, 1991.

[9] S. Olesen, J. Plogsties, P. Minnaar, F. Christensen, and H. Møller, "An improved mls measurement system for acquiring room impulse responses," in *Proceedings of NORSIG 2000*, (Kolmården, Sweden), pp. 117–120, IEEE Nordic Signal Processing Symposium, 2000.

[10] B&K, "Microphone handbook vol.1: Theory," tech. rep., Brüel & Kjær, 1996.

[11] V. Algazi, "Documentation for the UCD HRIR files," tech. rep., CIPIC Interface Laboratory - University of California at Davis, 1998.

[12] J. Sandvad and D. Hammershøi, "Binaural Auralization. Comparison of FIR and IIR Filter Representation of HIRs," in *Proceedings of the 96th AES Convention*, 1994.

[13] ISO, "266:1997 acoustics - preferred frequencies,"

[14] M. O. Hawksford, "Digital signal processing tools for loudspeaker evaluation and discrete-time crossover design," *Journal of the Audio Engineering Society*, vol. 45, no. 1-2, pp. 37–62, 1997.

[15] A. V. Oppenheim and R. W. Schafer, *Discrete-Time Signal Processing*. Prentice Hall, 1989.